Digital Connection

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History in the Making: Transatlantic DV

Originally in CQ Magazine, April 2003

"Kilo Foxtrot Six Delta XRay, this is Foxtrot Eight Kilo Golf Golf." So began the world's first transatlantic Amateur Radio contact using digital voice. On the morning of November 22, 2002 history was made. Digital HF voice became a reality.

Digital Voice (DV) is nothing new. Last year, Rich Moseson W2VU and I spent an afternoon playing with Alinco's then-new digital voice option on their FM handhelds. Before that, experiments with DV took place on the German high-speed packet network. Your digital cell phone uses digital voice. But these applications, and others like it, use plenty of bandwidth. With this accomplishment, the relatively strict bandwidth limitations of the HF bands were maintained.

First, let's take a closer look at what exactly was accomplished just before last Thanksgiving, along with a brief look at the technology that was used. Then, let's think about some of the implications for Amateur Radio.

Figure 1: Doug Smith, KF6DX, in the Ten-Tec shack. This was the USA side of the world's first transatlantic digital voice contact on HF radio, using unmodified Ten-Tec transcievers and Thales Communications' Skywave software. Photo by Eric Guinn, AC4LS

A few weeks ago, I spent some time speaking with Doug Smith, KF6DX, one of the participants in that historic contact. What follows is based upon that conversation.

Contact

It wasn't the first try. Since late summer, Amateurs in Paris and eastern Tennessee had been trying to get a new digital voice mode to work over transatlantic distances. They had been having some success, with words and phrases being copied, but it wasn't until that Friday in November that both callsigns and signal reports were exchanged, making it official. According to Doug, signals were S-5 to S-7, but the voice coming out of the speaker was almost as clean as an FM contact.

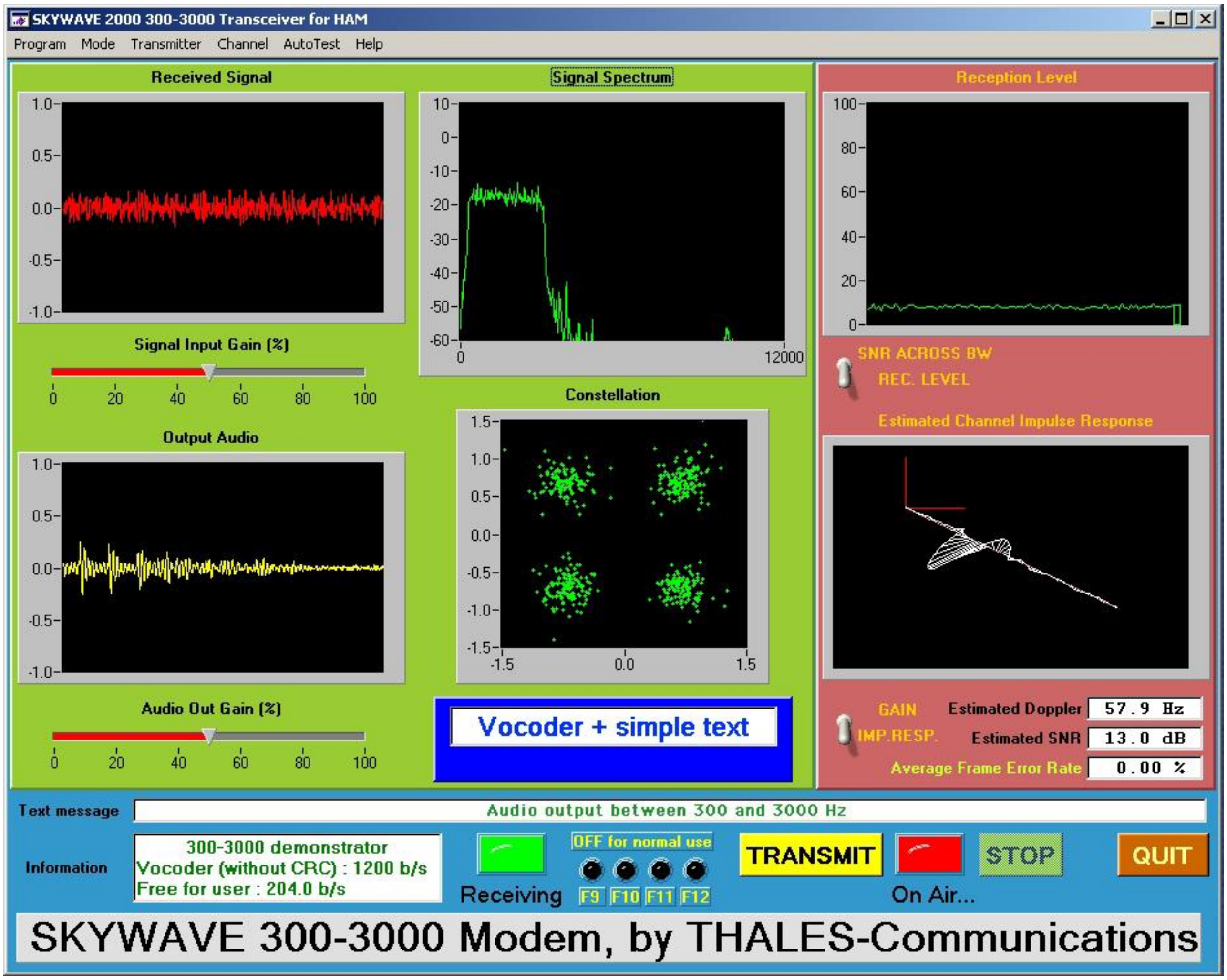
The contact between KF6DX and Didier Chulot, F5MJN (operating as F6KGG) was made with unmodified TenTec equipment and PC sound card software. Both sides were running about 100 watts with modest gain antennas. Doug told me that most HF rigs can handle the Digital Voice signal - all that's really necessary is decent frequency response from about 300 Hz up to 3 kHz. The bandwidth can even be a bit narrower, at the expense of requiring a higher signal to noise ratio.

When I asked Doug to describe the audio, he said that it has an FM-like quality to it. He explained that the digital decoder ignores any QRM or QRN, giving the audio a relatively noise-free sound. According to subjective testing that was performed, the audio was rated with an MOS (Mean Opinion Score) of 3.5, better than the toll-quality (like a telephone) standard of 3.0. Unlike SSB or AM, Digital Voice doesn't degrade gracefully when signal conditions deteriorate. Basically, you either hear it well or not at all. Doug's tests showed that you need a signal to noise ratio of about 6 to 10 dB, similar to what is typically needed for SSB.

The Software

The Skywave digital audio software, written by Thales Communications of France, requires a

Figure 2: A screen shot of Thales Communications' Skywave software. This application, which should become available later this year, was used for the first transatlantic digital voice contact late last year. The graphical features of the software allow the operator to monitor the performance of the radio system



modern PC and sound card for encoding and decoding the Digital Voice. Doug told me that we can expect the software to be released to the Amateur community some time this year. Although the price has not yet been determined, it should cost less than a decent antenna rotator or HF tribander.

Thales Communications SA of France is a company providing equipment to the shortwave broadcast industry, and a member of the DRM (Digital Radio Mondiale) consortium. It appears to me that the people at Thales recognized that there is a significant market in military and commercial HF radio users, so they developed a modified version of the DRM standard slightly. The most important modification, according to the description of the mode in the Jan/Feb issue of QEX, was a decrease in bandwidth, to allow the signal to fit into the 3 kHz bandwidth so common on SSB channels. They also selected a vocoder optimized for voice, simplified some of the control functions, and enhanced robustness and error detection.

It didn't hurt that some of the people at Thales happen to be hams. The DRM web site notes that DRM is still in the testing phase, and they are actively soliciting Radio Amateurs to help with their reception testing. It is very encouraging to see that the value of Ham radio is recognized internationally, and that fact wasn't lost on Thales. About a year and a half ago, they contacted KF6DX for some help with testing their HF version of DRM.

Doug is the head of the ARRL's Digital Voice Working Group (DVWG), which is part of the Technology Task Force. If Doug's name sounds familiar, it's probably because he is also well-known as the editor of QEX, as well as a senior design engineer for Ten-Tec. It was from the Ten-Tec offices <www.tentec.com> that the

historic contact was made, with Doug in his role as head of the DVWG.

Technical Details

It all started a few years ago, when a consortium of national and international radio broadcasting authorities, broadcast equipment manufacturers, and broadcasters finally decided on a standard for digital SW broadcasting. The Digital Radio Mondiale (www.drm.org) standard was accepted by the ITU, and work could begin on implementing it. DRM is an intelligent and flexible approach to moving shortwave broadcasting into the 21st century.

Briefly, the DRM standard calls for an up-to-10-kHz-wide digitally-encoded signal, made up of dozens of relatively narrow carriers, kind of like 100 PSK31 signals all crammed together. Sophisticated coding techniques are used to keep these closely spaced carriers from interfering with each other. In the Amateur version, a data rate of about 3600 bits per second can be maintained - better than 1 bit per second per Hz.

The basic signal, which for the broadcast standard is 4.5 kHz wide, carries not only the program information (voice or music, usually) but also information to control the receiver (which decoding scheme to use, etc.) and program content information (artist, title, upcoming programs, etc.). Most broadcasters will use wider-bandwidth versions of the DRM standard, but the program and control information will still reside in the relatively narrow basic carrier. The encoding scheme can be changed, even on the fly, for greater resistance to interference, with the receiver able to track these changes instantly. Broadcasters with wider bandwidth available will use it to improve signal fidelity as well as robustness.

One nifty trick DRM can use is a superposition of a compatible AM signal atop the Digital signal, so that listeners with traditional analog radios and those with digital radios can both hear the programming on the same channel. Another trick to increase the perceived audio fidelity is Spectral Band Replication (SBR). The high-frequency portions of an audio signal are mostly noise-like, kind of a hissing sound. SBR encodes the loudness and duration of these sounds, and at the receiver, the SBR decoder just kind of hisses at the right moments. The effect is amazing: Your mind really hears a much wider audio range than is actually present. To experience this for yourself, try the audio samples at the DRM web site.

Figure 3: With a few minor modifications, even my Heathkit SB-102 transciever can be made to handle Digital Voice. The most critical modification involves replacing the too-narrow 2.1 kHz SSB filter with one having at least 2.5 kHz, ideally more. The Skywave Digital Voice software prefers a 3 kHz audio bandwidth - most newer rigs are OK - although a slightly narrower bandwidth can be used, if you have a high enough signal to noise ratio.

To Probe Further

If you're interested in more of the technical details, there are a few resources you should review. The most detailed resource is the DRM standard itself (where most of this information came from), available from the EDSI web site <www.edsi.org> (search on "DRM"). There's a link to the EDSI site from the DRM site <www.drm.org>. The DRM web site is full of information, including audio samples, and even a place for Amateurs to sign up to participate in reception testing. (It is gratifying to see that DRM places a high value on the technical expertise of the Amateur community).

Of course, you should also have a look at the article by Cédric Demeure and Pierre-André Laurent in the January/February edition of QEX.

(QEX is the technical journal of the ARRL<www.arrl.org/qex>). Unlike EDSI and DRM, the article also explains the adaptations that were done for the Amateur and HF version of the system. Some of the modifications to the DRM standard include changes to accommodate the PTT (Push To Talk) nature of HF communications, the reduction in signal bandwidth, selection of a voice-only digital encoder, and removal or simplification of some of the features in which mainly broadcasters would be interested.

Lastly, you can take a look at the latest edition of TAPR's Packet Status Register, available on the TAPR web site <www.tapr.org/PSR>. A short article (written by yours truly) attempts to explain the DRM standard and it's Amateur cousin in simple terms. Although I admit that some of the explanations are not precisely accurate, the liberties taken are not significant, and they make it a little easier to conceptualize how the Digital Voice system works. Read it, then the QEX article, and then review the DRM web site, and you'll end up with a good understanding of the really ingenious technology at work here.

What does it mean?

The digital revolution started quite some years ago, but it's been picking up steam in recent years. Witness the home computer, Packet radio, and Al Gore's Internet. It's kind of exciting to realize that we're at just the beginning. Information transfer is the next great leap for humanity, and we're in at the ground floor.

What does this historic contact mean for Amateur Radio? Well, one of the first things that comes to mind is that we now have a new mode to play around with. We can start learning about Digital Voice, its advantages and disadvantages. This will keep us occupied for a little while, but there's much more than meets the eye. Some of the more subtle possibilities bear investigation.

One of the features of this Digital Voice system is that the specific encoding scheme can be changed on the fly. The detailed information required by the receiver is encoded into the signal in a standardized format, so the receiver can also switch modes at any time, automatically directed by the transmitting station. Say the signal is getting weak, as determined by an increase in the Bit Error Rate (BER). The receiving station can let the transmitting station know, and various actions can be taken. For example, the transmitter can increase power, or switch to a more robust encoding scheme. Perhaps the entire contact can move to a more favorable frequency, or even a different band - all auotmatically, all on the fly, all completely transparent to the users.

The control of the receiver by the transmitting station brings up some interesting possibilities. Trunking-radio type frequency usage. Enhanced spectrum utilization. Received audio quality feedback. Automatic power control.

The Digital Voice software can also pass small amounts of data, piggybacking on the main signal. What's envisioned is perhaps a few hundred bits per second. While that's not enough to send large files, it would be trivial to encode your callsign, or your QSL information. During a contest, you might actually log yourself in the other station's log! (Hopefully, they'll do the same). Now that would simplify logging quite a bit, wouldn't it?

The auxiliary data stream has other possibilities as well. Doug Smith, KF6DX offered a few examples in a recent conversation. "You could send a directed CQ, using [the auxiliary data stream] to include your areas of interest." Want to talk about Antennas or feedline? Just include that message in your signal. "You can also use it to send a file or SSTV image. The Slow Scan people won't even have to stop talking to send an image." That's because the digital voice signal and auxiliary data stream are sent at the same time.

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Now, I can just hear hundreds of digital-savvy readers saying to themselves "Gee, if I didn't send any voice at all, then I could use the whole signal for data. That would sure be a lot faster than SSTV or even 300 Baud Packet". Yes, it sure would be faster, but it would violate Part 97.307(f)(3), which specifies that the Baud rate of digital signals below 28.0 MHz must not exceed 300 Bauds.

OK, I can now hear all of you saying "hey, that means Digital Voice isn't legal on HF, since it is much faster than 300 Bauds". Nope. The FCC has traditionally looked at signal content as opposed to encoding method. In this case, Digital Voice is a Phone emission, and isn't subject to a data rate limitation. Sure, the supplemental data discussed above has a limit, but not the Voice signal (other than the 3 kHz bandwidth, of course). If you doubt this interpretation, ask yourself if it is legal to transmit (digitized) music via packet. No, it clearly is not legal, which in my opinion demonstrates that the content of the signal is what is important, not necessarily the mode.

What's new is Old

Digital Voice over HF is a new old idea. While digital voice isn't new, getting a better-than-toll-

quality voice signal into a 3 kHz HF channel is extraordinary. Just a few months ago, the first transatlantic HF contact using Digital Voice was made. Using a modified version of the Digital Radio Mondiale broadcast standard, Amateurs will soon have an extraordinary new mode available, the possibilities of which are only just starting to be explored. The rise of the Gigahertz PC processor, along with the ubiquitous Sound Card, has shown its power to the Amateur community for some years now. This new mode, while still experimental, shows that the power of these tools is only limited by the imagination.

Can you imagine other uses or advantages for digital voice, or the auxiliary data stream, that I haven't mentioned? I've touched upon just a few ideas, admittedly the more obvious ones. With the huge diversity of the CQ readership, there must be literally thousands of new and useful ideas on how this technology can be used. Please e-mail or write to me, even if you're not sure about your idea, and we can discuss it. I'll use the most interesting ideas in a future column.

In the February issue, I mentioned that I would offer an overview of the various HF modes this month. Obviously, that didn't happen. At the last minute, I decided that the historic and exciting Digital Voice contact last year would be more interesting, and I hope you agree. To get ourselves back on track again, in the June issue we will indeed take a look at the various HF digital modes.

Until then, 73 - N2IRZ.

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Sidebar: Do you need software?

Here's an interesting offer: A reader, who happens to be a software professional and Ham, is looking for a software project. It should be something useful to a wide audience, and of course be related to Amateur Radio. It also shouldn't be a re-hash or minor change to something that already exists, like another logging program. No sense reinventing the wheel.

This reader has considerable experience programming in DOS and Windows, and is also interested in learning more about Linux. In the amateur world, he has written a moderately popular contest logger, which is still in use, as well as dabbling in satellite trackers, propagation predictors and test study programs.

To avoid being inundated with requests, he prefers to remain anonymous for now. If you have any ideas for software you'd like to see, and have the commitment to work with a software developer to define what the software should do, please drop me a note and we can discuss it.